10 Intro to Digital Communications

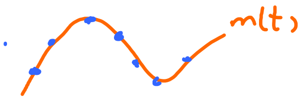
Friday, September 14, 2012
11:31 AM

10.1 Digitization and PCM [C&C Sec 12.1]

- Sampling
- Quantization

can use PAM to transmit this.

① $m(t) \rightarrow \boxed{\text{Sampler}} \rightarrow m[n]$



still analog ("any" real number is possible)

there are uncountably many of these.

② $m[n] \rightarrow \boxed{\text{quantizer}} \rightarrow m_q[n]$

digital (only a finite possible value.)

- may be sent directly via PAM
- may be encoded further
  Ex. converted to bits and then sent via PAM

$\Rightarrow$ PCM (pulse-code modulation)

digital PAM

This is used to emphasize the fact that $m_q[n]$ is digital

Recall that PAM is simply $\sum_n m[n] p(t - nT)$

10.2 Digital PAM

Here, $m[n]$ is digital

$\in \mathcal{A}$ ← alphabet

a set of M discrete values (symbols)

Ex. $\mathcal{A} = \{0, 1\}$

$\mathcal{A} = \{1, -1\}$

$\mathcal{A} = \{5, -5\}$

$\mathcal{A} = \{-3, -1, 1, 3\}$

There are many rectangular-based pulses that are used as $p(t)$ in digital PAM.

The use of these are referred to as "line codes". [See slide]

(Of course, you may replace the rectangular pulse with other pulse shape as well.)

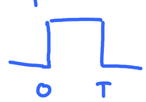
Observe that if we are given a digital message, e.g. 1011 0100 ...

we may not transmit it using pulse with 2 levels ($\pm A$)

it is also possible to group k bits and transmit each group using 2^k levels.

10.3 Digital PAM with noise

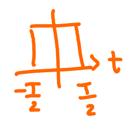
PAM: $x(t) = \sum_n \underbrace{m[n]} p(t - nT)$



← here, the pulse used is rectangular

again this is digital

note that the pulse is not



because we also require causality.

this is transmitted.

At the receiver, we will assume that the received signal is corrupted by the noise:

$$y(t) = x(t) + \underbrace{N(t)}_{\text{noise}}$$

this is generated in MATLAB by

$\underbrace{\sigma}_{\text{adjust the std.}} \times \underbrace{\text{randn}}_{\text{create the standard Gaussian noise}}$

We will consider two techniques to recover $m[n]$ back.

The first technique simply read the sampled values of $y(t)$.

The second technique uses "matched filter".

① Direct Sampling.

Let $r[n] = y(nT + \frac{T}{2})$. If $r[n] \geq 0$, set $\hat{m}[n] = 1$

$r[n] < 0$, set $\hat{m}[n] = -1$

Block diagram

Remarks: ① The "hat" in $\hat{m}[n]$ denote the fact that the recovered message may not be the same as the original message $m[n]$.

② In Section 9, you may recall that we sample at nT to recover the message.

Here, at the receiver, the sampling instants are shifted to $nT + \frac{T}{2}$ so that they are not at the boundary of the pulses.
(You may think of this as the same system discussed in Section 9 but with the time origin shifted by $\frac{T}{2}$.)

③ Without the noise, $r[n] = m[n]$.

④ $r[n]$ is used instead of $y[n]$ because

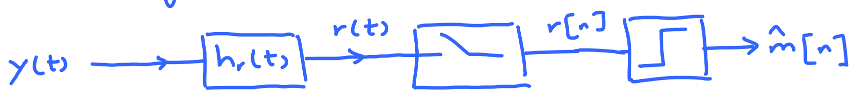
i) $y[n]$ suggests $y(nT)$ which is not the same as $r[n]$ and

ii) $r[n]$ suggests that further processing can be applied before $y(t)$ is sampled. This is more obvious in the second technique below.

In the slides, observe that the noise can introduce decoding error ($\hat{m}[n] \neq m[n]$.)

② To improve the performance of the system, we introduce one more processing at the receiver.
In particular, instead of sampling the received signal $y(t)$ directly, we will pass it throug a receiving filter $h_r(t)$ first.

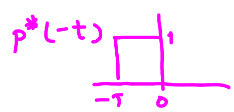
So, the block diagram of this receiver will be

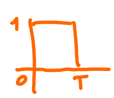


In more advanced class (e.g. digital commu.), we will see that the best $h_r(t)$ to use here is the matched filter

$$h_r(t) = p^*(T-t) = p^*(-(t-T))$$

complex conjugate.

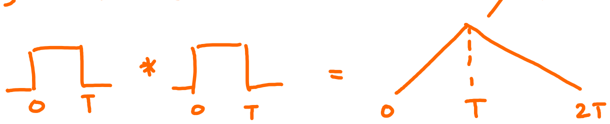


With $p(t) =$ 

, turn out that $h_r(t) = p(t)$

and the sampling instants are $nT + T$ instead of $\frac{T}{2}$.

In the slide, we see some motivation why the "$+T$" is there:



the peak of the triangle happens at T

The conversion from $r[n]$ to $\hat{m}[n]$ is the same as in the first technique.

Observe that with the use of $h_r(t)$, the whole pulse is involved in the recovering process, not just a sample from one position on the pulse.

10.4 Digital Modulation

Back to $A\cos(2\pi f_c t + \phi)$ as carrier

AM FM PM (when the message is $m(t)$)

ASK FSK PSK (when the message is digital)

shift keying

use finite number of amplitudes